IN THE SPECIFICATION

Please amend the paragraph beginning at page 7, line 3, as follows:

In line with a further preferred embodiment of the present invention, at least some of the data streams or data packets are temporarily buffered in the reproduction units before reproduction, with audio files typically involving buffering in the ~~region~~ range of approximately 1 to 5 sec. Preferably and with great advantage, e.g. in the case of realtime voice applications, the buffering is performed dynamically and so as to be matched to the circumstances of the network. The smaller the buffers, the shorter the latency for which it is necessary to wait before a stream can be played. Accordingly, it is advantageous to use the smallest possible buffers. The higher the quality of the network used, the smaller the buffers can be made, since in this case fewer failures occur and accordingly also fewer repetitions are necessary. Dynamic allocation of the buffer takes optimum account of this circumstance and can accordingly be used to optimize the latency. This buffering, which preferably takes place in a "ring buffer", firstly permits accurate synchronization, because the output pointer on the master and on the slave is simply set to be the same, and secondly this also makes correction mechanisms ("retry protocols") much more easily possible, which is of great significance within the context of the data integrity that is the aim here.

Please amend the paragraph beginning at page 7, line 30, as follows:

Particularly within the context of the output of audio files, it is found to be advantageous to design the synchronization of the individual reproduction units to be in the ~~region~~ range below 100 ms. Preferably, the propagation-time differences should be less than 10 ms or less than 2 ms, particularly preferably less than 1 ms. From psychoacoustics, it is known that normal hearing is capable of perceiving relatively large propagation-time differences of greater than 30 ms as echo, which is precisely what needs to be prevented

within the context of this invention. It is found that in the aforementioned "multichannel" mode, too, an accuracy in the ~~region~~ range of 1 ms is sufficient. The synchronization of data streams to this accuracy can no longer be assured in a typical network without active synchronization of the individual reproduction units, and in particular it is not possible simply to switch in further stations without active synchronization. Typically, the network is a conventional, wired network, but it may preferably also be a wireless network, particularly a radio network (e.g. Wifi, wireless fidelity, also called IEEE802.11b, or follow-up standards at a higher data rate, such as IEEE802.11a). If, as proposed in line with another preferred embodiment of the present invention, it is necessary to switch in a further reproduction unit synchronously, then this is preferably done by virtue of the unit which has been switched in automatically aligning itself with the present master and starting reproduction itself after buffering some of the data. Similarly, it is sometimes found to be very advantageous to be able to set the delay of a client in specific fashion relative to the master. This means that large spaces, churches etc. can be supplied with data much better acoustically and can compensate for acoustic properties/propagation times in these buildings. This then involves a deliberate, i.e. desired and systematic, delay, however. The time shift used for this when reproducing on different units is set to be constant and remains synchronized.

Please amend the paragraph beginning at page 12, line 29, as follows:

This maximum network delay time can be ascertained from time to time by the master, for example, and can then be stored. Typically, for the sake of security, this maximum network delay time can have an additional contribution added to it, for example in the ~~region~~ range from 1-20 ms. This additional contribution serves to cover the network transmission tolerance (safety margin) and the processing time in the transmitter and receiver. This maximum network delay time should be adapted dynamically, since this maximum

network delay time can change greatly on account of alterations in the network. This characteristic time thus changes, for example when the load on the network rises (typically an increase in the maximum network delay time), or when further reproduction units or other units are switched in or disconnected.

Please amend the paragraph beginning at page 16, line 4, as follows:

As an exemplary embodiment of the present invention, a system will be described in which a "transmission unit" distributes a continuous data stream delivered by a (digital or analog) audio data source wirelessly over a plurality of distributed reproduction units (typically active loudspeakers), with the latter decoding and outputting various channels of the same data stream. To this end, the transmitter unit has a CPU, i.e. a processor, buffer store, and at least one bidirectional communication interface, in the example described an 802.11b radio network interface, and an audio input for analog or digital audio data and also its own audio output (that is to say that it is also a reproduction unit). The other reproduction units use the same architecture, but instead of an audio input have a digital and/or analog audio output and possibly power amplifiers and sound transducers/loudspeakers 13.

Please amend the paragraph beginning at page 24, line 18, as follows:

When a data packet is lost, this typically occurs on the network and normally for a plurality of or all of the slaves in the "channel". To avoid unnecessary loading of the network, an intelligent retry method is used. All of the slaves receive the "new" data packet at approximately the same time and are therefore able to establish approximately simultaneously that data have been lost. Each client now delays an individual time (derived from the IP address or MAC address under random control or by means of an algorithm), in the ~~region~~ range of 1 to, by way of example, 30 ms, before it transmits a "retry request". This

retry request is then sent by broadcast to the UDP port which is specific to the "channel", and can thus be received by all of the stations which are connected to the channel – not just by the master. While the clients wishing to initiate a retry wait for the individual time, reception of UDP datagrams is continued. A retry initiated by another client ends the waiting and prevents the client's own retry from being sent if it is the same (the same first byte address which is not present) or is a request for even more data – this effectively prevents multiple identical retry requests and minimizes the load on the network.

Please add the following beginning at page 28, line 15, as follows:

13 Loudspeaker